Am79C02/03/031 DSLAC™ Devices Adaptive Balance Feature

Advanced Micro Devices

Application Note

The cause of four-wire echo in subscriber linecards is the mismatch of the two-wire impedances and the echo balance circuit. Because of the mismatch, much of the energy from the received signal is returned in the transmit direction. The conventional method of removing this echo is to add an echo balance signal in the transmit path while terminating the two-wire circuit with a compromise network. The compromise termination network is the result of extensive loop surveys representing the average impedance characteristics of a particular nation's subscriber local loops.

The Am79C02/03/031 DSLAC™ devices use an advanced adaptive filtering technology to remove the echo energy from the transmitted signal. The adaptive balance filter in the DSLAC devices dynamically sets coefficients of the digital balance filter (B filter) to maintain the best echo cancellation under changing local loop conditions.

A significant improvement of the adaptive balance performance has been achieved with the DSLAC devices, especially during double-talker conditions and tone transmission. The DSLAC devices include a decorrelation controller, a digital pre-balance, and a low-level signal detector.

This application note is intended to present guidelines that should be followed in order to achieve a satisfactory performance in the Continuous Adapt or the Adapt and Freeze mode of the DSLAC devices. To effectively use the applications shown in this document, the user must be familiar with the DSLAC device data sheet and the AmSLAC2™ software.

A summary of DSLAC device commands affecting adaptive balance is included in Appendix 1. There are also five programmable parameters that control the adaptive balance, i.e., Echo Path Gain (EPG), Error Level Threshold (ELT), Decorrelation Threshold (DCR), Digital Pre-Balance (DPB), and Low Signal Detection Threshold (LST). The hexadecimal values of these parameters may be obtained from Appendix 2.

ADAPT AND FREEZE MODE

The Adapt and Freeze mode is intended for use in applications such as voice mail systems or a PABX, where the loop condition and subscriber terminal equipment are static. It can also be used in central office applications on a per line basis during installation or maintenance, or on a per call basis if a training signal at the beginning of a call is considered acceptable.

For this application, the telephone must be off-hook and a white noise training signal must be sent to the receive side of the DSLAC device to train the B filter prior to freezing the coefficients. At the beginning of the training sequence, the Adaptive Balance Filter (ABF) mode bit should be set to 1 to enable the Adaptive mode. The same bit should be set to 0 for the end of the training sequence. Once the adaptation is complete, the coefficients can be read out of the DSLAC device and stored in the system configuration record so that the adaptation need only be done when the system changes.

Adapt and Freeze: Training Signal

A white noise input signal with a flat frequency response between 300 Hz and 3400 Hz should be used for training purposes. Individual tones and narrow-band noise should be avoided. Noise with a flat response across the voice band will result in the best echo balance performance across the telephony voice band while a tone or narrow band noise may enhance some frequencies at the expense of the others. A digital recorded announcement could be used. White noise is available from W&G PCM-4 for laboratory tests. Sources of flat response noise inside an exchange could be in the tone or maintenance module.

The level of the white noise should not exceed –10 dBm0 to avoid clipping at the input of the A/D stage when the noise has a Gaussian distribution and the loop condition is not favorable.

Adapt and Freeze: Training Duration

The training duration should not be less than 100 ms. A maximum of 500 ms training duration is recommended.

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Adapt and Freeze: Initial Coefficient of B Filter

It should be noted that not all B-filter coefficients are valid to initiate adaptation. For the FIR portion of the B-filter coefficients, the first 12 bytes should be initialized to a valid zero, which is Hex2AF2AF2AF2AF2AF2AF-2AF2AF. For the IIR portion of the B filter, the last two bytes should be initialized to a value of 0A80 Hex, which is equivalent to a decimal +0.5. A negative IIR coefficient is prohibited. A positive IIR coefficient with values higher than 0.95 may cause oscillation. Other seed values (FIR portion) used to speed up the convergence of adaptation may be obtained by reading back the coefficients after adaptation to a nominal load or a compromise network.

Adapt and Freeze: Echo Path Gain (EPG)

Since there is no double-talker condition during the B-filter training period, the EPG should be programmed to a maximum value Hex 0000, which is equivalent to +14 dB. This will permit the B filter to fully adapt to the training signal.

Adapt and Freeze: Error Level Threshold (ELT)

The ELT should be set to Hex 75, which is equivalent to –30 dB for optimum operation.

Adapt and Freeze: Decorrelation 1/2, Pre-Balance and Low-Level Signal Detect

These three parameters are for continuous adaptation only. They should be disabled for the adapt and freeze case.

Adapt and Freeze: Testing the Result

Because there may be some variation in the balance achieved, it is recommended that the echo balance be measured. If it is not satisfactory, change the ELT and rerun the adaptation.

CONTINUOUS ADAPT MODE

The Continuous Adapt mode is intended for use in applications where local loop conditions and customer premises equipment are changed frequently and a high echo return loss needs to be maintained. In this mode, the adaptive B filter will be turned on from the beginning of a call and will stay on until the call ends. To activate the adaptive balance, a B filter turn-on command needs to be sent, followed by an enable adaptive B-filter (ABF) command (see Appendix 1). The receive signal to the B filter will be voice, tones, music, background noise, fax/modem signals, or combinations of the above.

The Continuous Adapt mode is implemented with an eight-tap digital B filter that automatically adjusts its coefficients to match the impulse response of the two-wire loop. The B-filter adaptation is based on a sign-based Least Mean Square algorithm that usually converges within 100 ms. Only the FIR part of the filter adapts, so the circuit is unconditionally stable.

An uncontrollable or excessively adapting B filter is undesirable due to the divergence of the filter coefficients. This is especially true for single tone excitation, which may cause the balance filter to converge then diverge. A residual error-level detection is provided in the DSLAC device by testing the ratio of the energy level of the error signal to the energy level in the receive side against a programmed error level threshold (ELT). The adaptation will be stopped when the ratio goes below the threshold.

In a double-talker condition, that is, when both near-end and far-end signals appear at the same time, the adaptation will be somewhat misguided because of the merging of the near-end signal and the echo. A conventional way to solve this problem is to compare the near-end with the far-end signal levels. If the ratio exceeds a predetermined threshold (Echo Path Gain, [EPG]), the adaptation will be stopped. Double-talker detection will not always be reliable because the near-end signal can be small enough not to trigger the double-talker condition, but large enough to disturb the adaptation. This causes a problem for high speed modem traffic, which requires a non-adaptive duplex (double-talker) connection. A continuously active adaptation in modem connections will result in high bit error rate or even connection hang-up due to the noise generated by the finite step size of the coefficient update.

A sign-based decorrelation controller is provided in the DSLAC device for the purpose of detecting a cross correlation between the far-end signal and echo residual. The same process is also true when a near-end signal is present at the analog front-end, because a near-end signal is always uncorrelated with a far-end signal in the actual application. Thus, the DSLAC device is able to stop the adaptation once a decorrelation is reached, with or without the presence of the near-end signal.

In addition to function as a double-talker detector, the EPG circuit is also used to improve the four-wire loop stability. See Continuous Adapt Mode: Echo Path Gain for details. A low-level signal detection (LST) is also provided in the device to inhibit adaptation when the far-end signal level is too small.

In summary, there are four conditions in which the DSLAC device will automatically stop adapting in the Continuous Adapt mode:

- When the residual error signal is below a programmed threshold ELT.
- When the correlation value is below a programmed threshold DCR.
- When the near-end signal exceeds a programmed threshold EPG.
- When the far-end signal is below a programmed threshold LST.

The following are guidelines that are essential to a satisfactory operation of continuous adaptation. The EPG, ELT, LST, or DCR values given in the following description are often given in dB or decimal. The tables in Appendix 2 must be used to convert them to the hexadecimal codes required by the Write EPG, the Write ELT, or the Write Adaptive B control commands.

Continuous Adapt Mode: Initial Coefficients for the B Filter

The first 12 bytes of the B filter (FIR portion) should be initialized to a Hex 2AF2AF2AF2AF2AF2AF2AF2AF, which is equivalent to a decimal zero. The last two bytes of the B filter (IIR portion) should be initialized to a value of Hex 0A80, which is equivalent to a decimal +0.5. A negative IIR coefficient is prohibited. A positive IIR coefficient with values higher than 0.95 may cause oscillation.

Other seed values (FIR portion) that are optimum for the operation can be obtained by reading back the coefficients after an off-line adaptation to a nominal load or a compromise network. In this case, the ELT, EPG, and DCR1/2 should be set to the maximum value for a full adaptation. A flat noise or conventional telephony noise (G.227) is recommended as an excitation source. The pre-balance should be turned on and a proper value is selected, see Continuous Adapt Mode: Digital Pre-Balance.

Continuous Adapt Mode: Echo Path Gain (EPG)

The following is the equation governing the double-talker detection and its relationship with EPG.

If Py(i) > EPG * Px(i), then adaptation stops

where: Px(i) and Py(i) are the short-term energy estimates of the far-end and near-end signals, respectively, at instant i. The equation states that double-talker is detected if the near-end signal power is greater than the EPG value times the far-end input power, and the adaptation should be stopped. Note that the near-end signal defined here may contain some of the returned echo. There is a high pass filter associated with Py so that a low-level power induction (50 Hz/60 Hz) won't inhibit the adaptation.

The choice of EPG is critical for proper adaptive balance operation. A compromise EPG value has to be selected between a high sensitivity (low EPG) and a low sensitivity (high EPG) of the double-talker detection.

An over-sensitive double-talker detection, which means a Low EPG value, will cause a returned echo to trigger a double-talker condition, especially with a high-impedance loop connected.

An under-sensitive double-talker detection, which means a High EPG value, will cause the B filter to attempt to cancel the near-end speech.

Therefore, a compromise EPG value should be selected. The compromise EPG value may still result in an unreliable double-talker detection because of the boundary where the near-end signal is small enough not to trigger the double-talker condition, but large enough to disturb the adaptation.

In addition, if a DSLAC device is connected to a non-adaptive far-end hybrid that has a poor transhybrid loss, there is a strong possibility that an alternating converging and diverging burst will occur when a near-end signal is sent. This is due to the fact that the near-end signal is echoed at the distant hybrid, turning into a far-end signal, and then the B filter tends to cancel the near-end signal resulting in a divergence. Selecting a proper value of EPG, which is usually less than the transhybrid loss (in dB) of the distant hybrid, will effectively stop this bursting loop instability.

As evident, it will be difficult to choose an EPG value to satisfy all three requirements as mentioned above: over-sensitivity, under-sensitivity, and bursting loop stability. Therefore, a pre-balance network has been added to reduce the echo by at least 6 dB, which alleviates the over-sensitivity problem and makes the EPG selection possible.

For a chosen value of EPG, return echo should not trigger double-talker detection with nominal load applied. The Digital Pre-balance that reduces the return echo by at least 6 dB is able to prevent any false double-talker detection. This pre-balance maintains near-end signal detection. The following equation is recommended for selecting a proper EPG value:

$$EPG = 20 \log \frac{G_{44o} + G_{44}}{2} + AR + AX$$

where: G_{44o} and G_{44} are open and short circuit gains respectively for the SLIC in decimal.

AR is the analog receive gain in dB.

AX is the analog transmit gain in dB.

Test all expected worst-case loop conditions to make sure the adaptation is not inhibited by the programmed EPG. A minor adjustment of the EPG is sometimes necessary. Turn on the pre-balance and program a proper value while testing the EPG.

Continuous Adapt Mode: Digital Pre-Balance (DPB)

The digital pre-balance (DPB) circuit is a one tap digital filter provided in the DSLAC device. The digital pre-balance is used to reduce echo in the transmit direction, allowing the use of a lower EPG value to solve bursting loop instability problems.

This is done by adding the fifth tap product of the B filter and a programmable coefficient (three-nibble CSD) with the decimator output. The coefficient should be selected such that a maximum echo cancellation is achieved with the B filter disabled and a nominal termination. A fine tuning of the coefficient is usually needed to get a flat echo cancellation between 200 Hz and 3400 Hz.

Since the pre-balance is a one-tap 16-kHz digital filter, a perfect cancellation is not possible and a frequency dependent performance is expected due to the digital delay. The optimum pre-balance value is generated by the AmSLAC2 software. If a pre-balance function is already implemented in the analog path (i.e., differential transformer drive), then the DPB should be disabled.

Continuous Adapt Mode: Error Level Threshold (ELT)

One of the conditions where the continuous adaptation process is stopped is when the error (echo cancel output) level is less than a predetermined fraction of echo cancellation. The equation shown below represents the relationship:

If Pe(i) < ELT * Px(i), then the adaptation stops

where: Pe(i) and Px(i) are the short term energy of the residual error and far-end signal respectively at instant i and ELT is the error level threshold.

Under steady-state, Pe(i) approaches Px(i)*(ELT+GR+R)

An ELT value of –30 dB is recommended for most applications. Since the far-end signal sensing point is located at the output of the R/GR block, the actual error (echo residual) level will be affected by GR gain block or R Filter. The reason for choosing the error sensing point at the output of the LPF is to filter out 50-Hz/60-Hz power induction components.

Continuous Adapt Mode: Decorrelation Controller (DCR1, DCR2)

A decorrelation controller is implemented in the DSLAC device for providing reliable operation under a double-talker condition, especially a modem connection. The decorrelation controller is designed to stop the adaptation once the echo residual is uncorrelated to the far-end signal with or without the presence of a near-end signal. The adaptation will be resumed when the correlation value again exceeds a threshold.

A unique sign-based correlation detection has been used in the decorrelation controller using two threshold values, DCR1 and DCR2. The selection of DCR1 and DCR2 depends upon the compromise between the adaptability to loop condition changes, which requires a lower DCR1/2 threshold, and the stable modem connection, which requires a higher threshold. A value of 0.22 for both DCR1 and DCR2 is recommended for most of the applications. DCR1 is an eight-tap correlator; DCR2 is a one-tap correlator that helps to improve the adaptability of the overall system under certain conditions, such as tone sweep, for example.

Continuous Adapt Mode: Low Signal Level Detection (LST)

A low signal level detector is provided to stop the adaptation when the far-end signal is below a certain threshold. This is intended to improve the transmission performance that may suffer from the adaptation noise in the A to A measurement.

The average amplitude of the far-end signal is compared to the product of a constant reference, 0 dBm0 and a three-nibble CSD coefficient, LST. Since the LST-test access point is located after the R Filter and GR blocks, the LST value should be adjusted accordingly. An LST value of –35 dB is recommended with 0 dB gain in GR and R blocks. Generally, an LST value should be selected by the equation:

$$LST = -35 - GR - R \text{ (in dB)}$$

Other Uses of Adaptive Balance

The primary reason for adaptive balance is to adjust the B filter for optimum four-wire echo performance, but some of the features of adaptive balance can be used to perform simple measurements. For instance, the EPG and ELT coefficients can be used together to obtain an indication of echo-level and four-wire echo performance. This can be done by using the table on page 10 to relate the point where the B filter starts to adapt as the ELT value is changed to find the threshold point to a dB value. Note also that the line impedance measurements can be made by comparing B-filter coefficients achieved by the use of adaptive balance in the field against calibration coefficients. This method can be used to measure line impedance in terms of magnitude or phase, or to discriminate between long or short lines to perhaps select between different sets of stored coefficients.

PRACTICAL RESULTS

To demonstrate the ability of adaptive balance to correct for degradation of transhybrid balance return loss (TBRL) with increasing line length, AmSLAC2 was used to provide the initial coefficients for a 600-Ω line termination with a zero length line. A line length simulator was used to generate the effective line length. The TBRL results for 0-km, 0.9-km, 1.8-km,. 2.7-km, and 3.6-km line lengths using 22-gauge wire with fixed B-filter coefficients are shown in Figure 1. The 0-km result is the highest, with performance degrading with the longer line lengths. Adaptation was then applied and the results were recorded. The adaptation used the white noise source from a W&G PCM-4, which was also used for measurements. Although the 0-km line had good results with the fixed coefficients, the use of adaptive balance improved the results significantly as shown in Figure 2. Figures 3, 4, 5, and 6 show the results for the other line lengths with adaptation applied to improve the results. Note that the 3.6-km line length, the worst of the group with less than 10 dB TBRL, improved to 24 dB or better over the voice band.

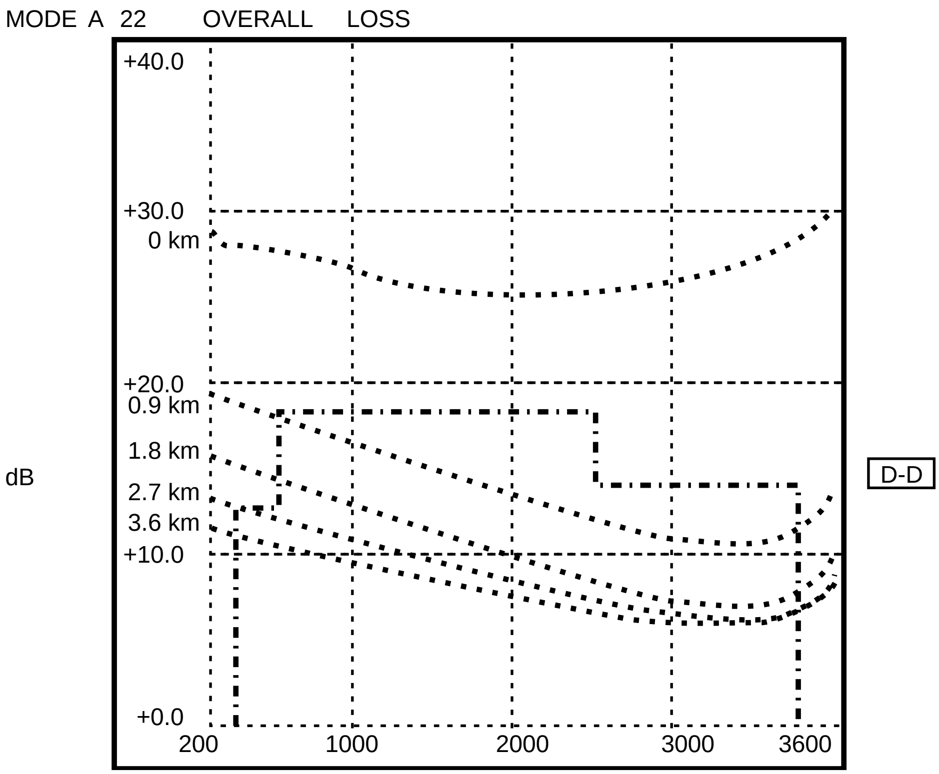

Figure 1. TBRL for Various Line Lengths

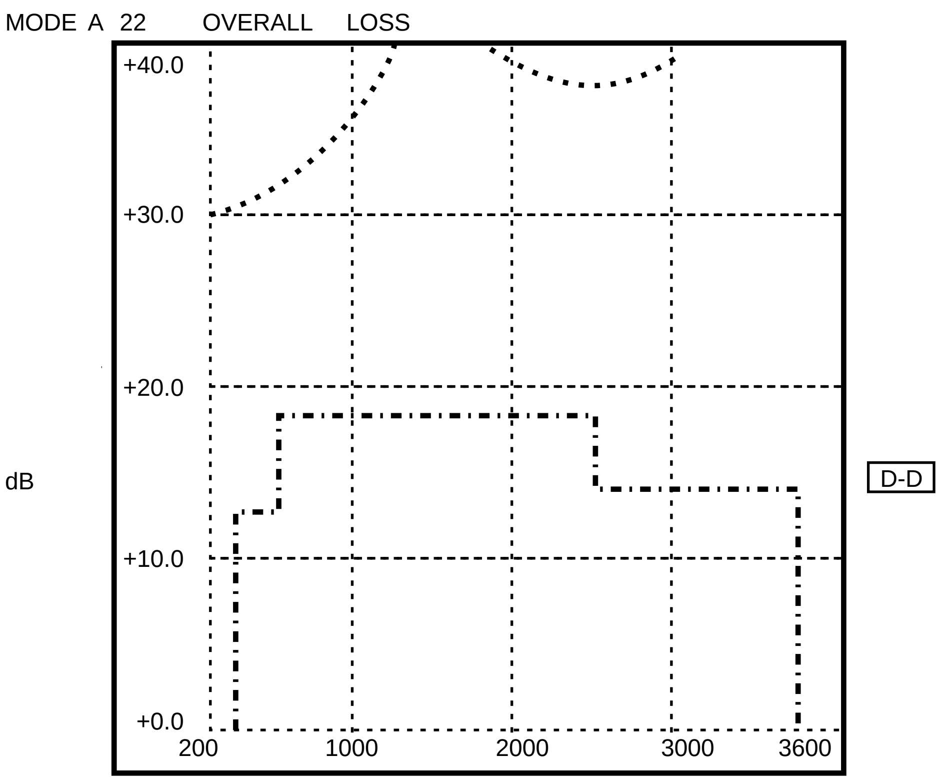

Figure 2. Adaptive Balance, Line of 0 km

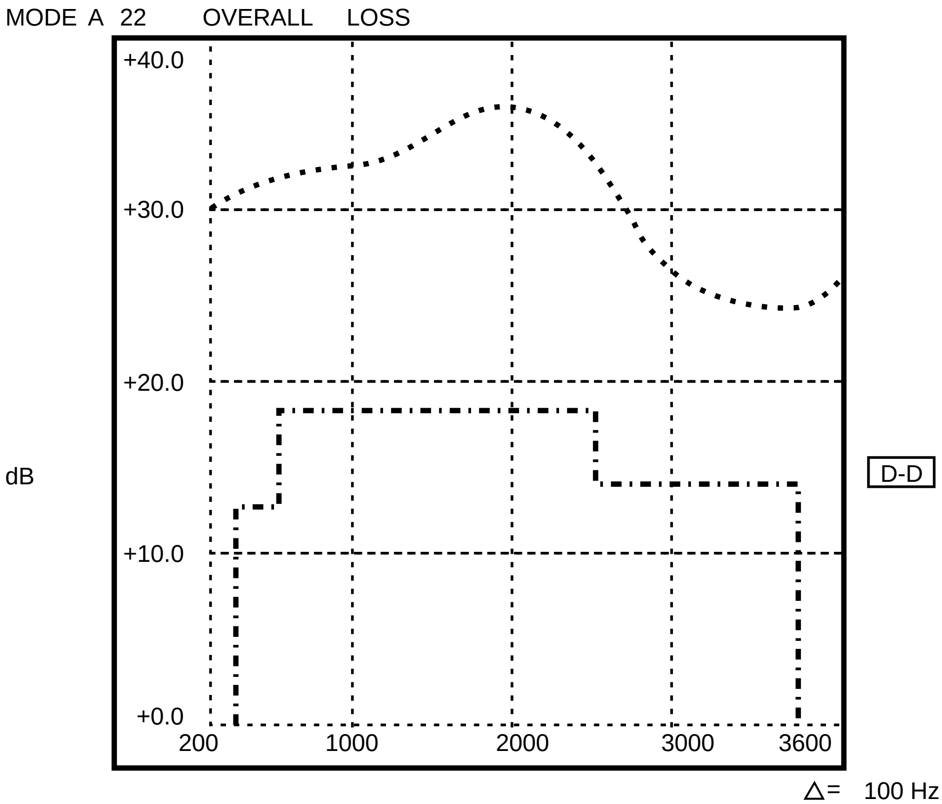

Figure 3. Adaptive Balance, Line of 0.9 km

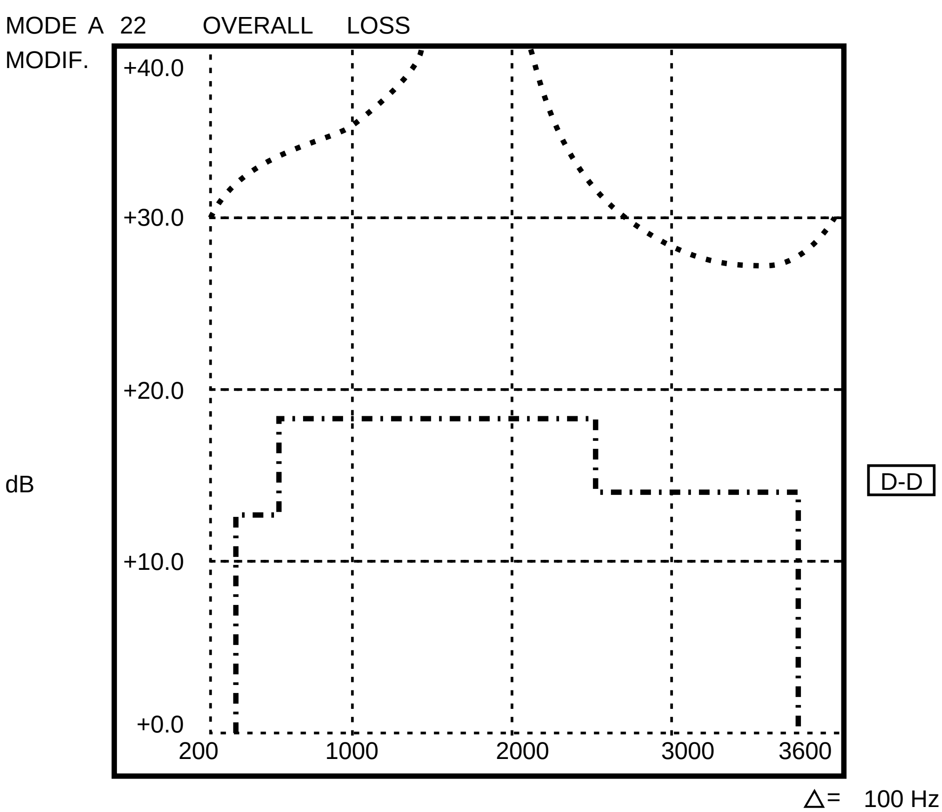

Figure 4. Adaptive Balance, Line of 1.8 km

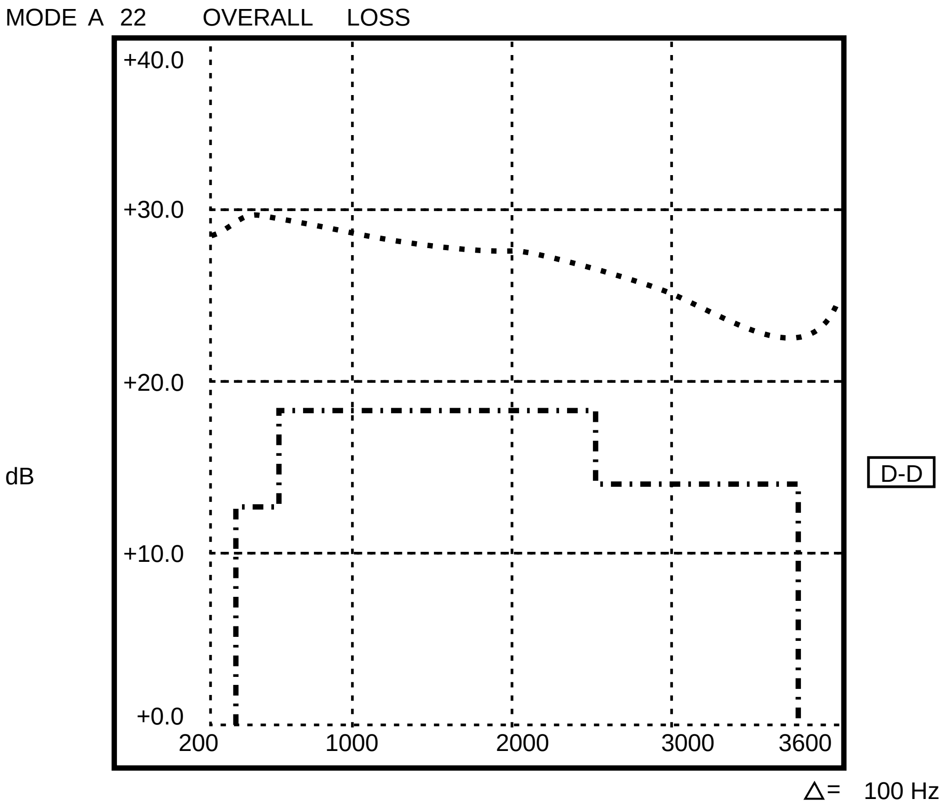

Figure 5. Adaptive Balance, Line of 2.7 km

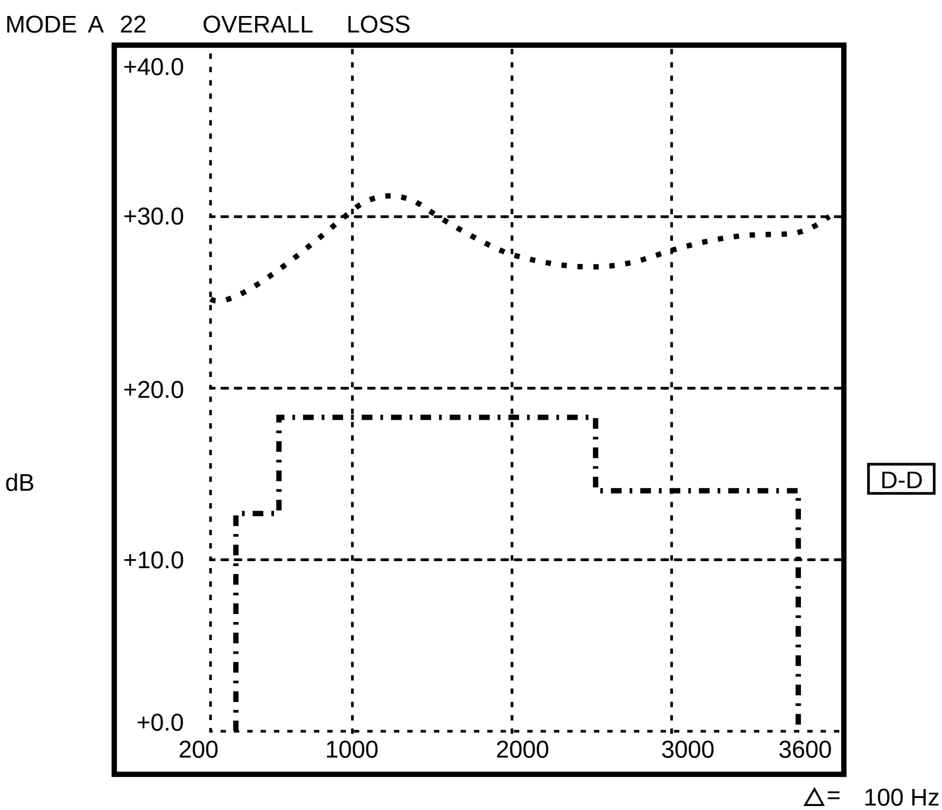

Figure 6. Adaptive Balance, Line of 3.6 km

Appendix 1: Summary of Commands Affecting Adaptive Balance

The following MPI (Microprocessor Interface) commands affect adaptive balance:

- Reset (Hex 02)

  This software reset command sets the DSLAC device to the Non-Adaptive mode.

- Write Operating Function (Hex 60)

  Bit D7 of the input data byte following the command is the Adaptive Balance Filter (ABF) mode bit. ABF = 1 sets the Adaptive Balance mode, while ABF = 0 clears the Adaptive Balance mode and freezes B-filter coefficients.

- Read Operating Conditions (Hex 61)

  This command reads the ABF mode bit defined above as bit D7 of the output byte.

- Write Echo Path Gain (Hex 8C)

  This command writes the Echo Path Gain (EPG) to the DSLAC device. The EPG controls the double-talker detection threshold level. Four input data bytes must follow the command.

- Read Echo Path Gain (Hex 8D)

  This command reads the EPG defined above.

- Write Error Level Threshold (8E)

  This command writes the Error Level Threshold (ELT) to the DSLAC device. The ELT controls the residual error signal level at which the adaptation will be turned off. One input data byte must follow the command.

- Read Error Level Threshold (8F)

  This command reads the ELT defined above.

- Write Adaptive B Filter Control Coefficients (90)

  This command writes the DCR1, DCR2, LST and Pre-balance values to the DSLAC device. DCR1/2 controls the decorrelation threshold, while LST controls low-level signal threshold. Five data bytes should follow the command with the first byte dedicated to DCR1, the second byte to DCR2, the third byte to LST, the fifth byte to Pre-balance, and the fourth byte to both LST and Pre-balance.

- Read Adaptive B Filter Control Coefficients (91)

  This command reads the control coefficients defined above.

- Write Adaptive B Operating Functions (64)

  This command writes the control bits of the DCR1, DCR2, LST, and Pre-balance to the DSLAC device. One data byte should follow the command with bit definition as follows:

  Bit 0 = 0 Digital pre-balance is disabled

  Bit 0 = 1 Digital pre-balance is enabled

  Bit 1 = 0 DCR1, DCR2, and LST are disabled

  Bit 1 = 1 DCR1, DCR2, and LST are enabled

- Read Adaptive B Operating Functions (65)

  This command reads the DCR1, DCR2, LST, and Pre-balance control bits as defined above.

Appendix 2: Hexadecimal Conversion Table

The following tables give a conversion between the dB or decimal and the Hex values with respect to ELT, EPG, DCR1/2, and LST. For those values not listed here, the user needs to refer to CSD formulas in the DSLAC device data sheet for a conversion.

Table 1. Error Level Threshold (ELT)

CSD (Hex)	dB value	CSD (Hex)	dB value
22	–10	A4	–26
32	–11	n/a	–27
91	–12	25	–28
B2	–13	35	–29
A2	–14	75	–30
n/a	–15	B5	–31
23	–16	A5	–32
33	–17	n/a	–33
92	–18	26	–34
B3	–19	36	–35
A3	–20	76	–36
n/a	–21	B6	–37
24	–22	A6	–38
34	–23	27	–40
74	–24	A7	–44
B4	–25	97	–48

Table 2. Echo Path Gain (EPG)

CSD (Hex)	dB value	CSD (Hex)	dB value
2329	–9	2AA1	3
AA29	–8	2A21	4
2B39	–7	3990	5
9888	–6	1AF0	6
8FB9	–5	1D20	7
A4A9	–4	1610	8
2A2A	–3	1B00	9
3AAA	–2	2300	10
3CAB	–1	3100	11
0800	0	0100	12
41E3	1	1000	13
3352	2	0000	14

Table 3. Decorrelation Control (DCR1, DCR2)

CSD (Hex)	Correlation	CSD(Hex)	Correlation
80	0	C2	0.23
A3	0.1	D2	0.24
B3	0.11	91	0.25
D3	0.12	52	0.26
53	0.13	42	0.27
33	0.14	32	0.28
23	0.16	22	0.31
A2	0.19	A1	0.38
B2	0.22	01	1.0

Table 4. Low Level Signal Detection (LST)

CSD (Hex)	dBm0, value	CSD (Hex)	dBm0, value
A65	–30	327	–40
AB5	–31	337	–41
AA5	–32	467	–42
3A5	–33	AB7	–43
326	–34	AA7	–44
336	–35	2A7	–45
C66	–36	n/a	–46
AB6	–37	n/a	–47
AA6	–38	n/a	–48
3A6	–39	n/a	–49

Notes:
1. Assuming GR = 0 dB and R = 0 dB.
2. n/a means not available.